```
admin> set system-ip-addr = 192.168.100.128
admin> write
IP-GLOBAL written
```

For MultiVoice, the `system-ip-addr` parameter may use the following IP addresses associated with a TAOS unit:

- The IP address of the shelf controller Ethernet port
- The IP address of an Ethernet-3 card Ethernet port
- The soft IP address associated with the TAOS unit

The IP address of both the shelf controller Ethernet port and the Ethernet-3 slot card are associated with physical network connections on the TAOS unit. The soft IP address is a virtual network connection, which is broadcast to the network. The soft IP address may be used to communicate with the TAOS unit so long as one of the unit's physical IP interfaces (for example, an Ethernet slot card port) is up.

Note When a APX is configured for redundancy, the `system-ip-addr` parameter should be set to the soft IP address. This soft IP interface always routes to the primary shelf controller and is hidden from the secondary. When failover occurs and the secondary controller becomes primary, the soft IP interface is re-initialized and is now associated with the new primary controller.

The soft IP address is an internal interface that never goes down and is always associated with the TAOS unit itself rather than a specific hardware interface. The soft IP interface is set up by the system once the TAOS unit's power is turned on and the shelf controller is up. The `ip-interface` profile with the zero index is reserved for the soft IP interface. For example:

```
admin> dir ip-interface
  6  06/17/1999 03:06:00    { { any-shelf any-slot 0 } 0 }
 19  06/21/1999 23:54:02    { { shelf-1 left-controller 1 } 0 }
 19  06/25/1999 17:45:30    { { shelf-1 right-controller 1 } 0 }
```

Note If RIP is enabled, a TAOS unit advertises the soft IP interface address as a host route (with a prefix length of /32) using the loopback interface. If RIP is not enabled, routers one hop away from the TAOS unit must have a static route to the soft interface address.

The soft IP interface is activated by entering an IP address for `{{ any-shelf any-slot 0} 0}`. The following example shows how to set the soft IP address. In this example, the address is set to `192.168.100.128/24`:

```
admin> read ip-interface { 0 0 0 }
IP-INTERFACE/{ { any-shelf any-slot 0 } 0 } read
admin> set ip-addr = 192.168.100.128/24
admin> write
IP-INTERFACE/{ { any-shelf any-slot 0 } 0 } written
```

For H.323 VoIP call processing, the `system-ip-addr` parameter identifies gateways to the gatekeeper and for call routing by the gatekeeper (see "Configuring MultiVoice Gateways" in the *MultiVoice Access Manager User's Guide*). When this parameter uses the soft IP address, the TAOS unit can continue to run keep-alive registration with the gatekeeper and accept calls even in the event the network connection on the shelf controller fails.

For IPCD VoIP processing, the TAOS unit can allocate its own system IP address for the listen IP address and Real-time Transport Protocol (RTP) port and can specify its

own send address and RTP port. When allowing a TAOS unit that is acting as MultiVoice Gateway to allocate its own address, you must set the `system-ip-addr` parameter in the `ip-global` profile to an interface address other than the shelf-controller Ethernet port. For example, the following commands set the system address to the address of a port on an Ethernet card in slot 12:

```
admin> get ip-interface { { 1 12 1 } 0} ip-address
[in IP-INTERFACE/{ { shelf-1 slot-12 1 } 0 }:ip-address]
ip-address = 1.1.1.1/24
admin> read ip-global
IP-GLOBAL read
admin> set system-ip-addr = 1.1.1.1/24
admin> write
IP-GLOBAL written
```

VoIP traffic (communications between TAOS unit) is processed through the Ethernet slot card interfaces on a TAOS unit by setting up either host routes or network routes for communication between remote gateways using IP addresses assigned to the Ethernet slot card interfaces on a TAOS unit. See "Configuring IP routing for H.323 call processing" on page 2-28 and "Configuring IP routing for IPDC call processing" on page 2-37 for details.

Detecting and responding to misdirected ICMP packets

For H.323 VoIP, the value of the `send-icmp-dest-unreachable` parameter in the `ip-global` profile should be set to yes for MultiVoice operations. The yes setting allows a TAOS unit unit to detect and respond to misdirected ICMP packets by responding with an ICMP-unreachable packet, rather than redirecting the packet to the shelf controller when operating under heavy call volumes. Enabling this parameter also reduces the load placed on the shelf controller.

For each VoIP call, UDP packets can arrive at a rate of 200 packets per second, per direction. When setting up or tearing down calls, if the TAOS unit unit is not listening on a port for these packets, it can generate ICMP packets at a rate of one ICMP packet per one UDP packet, which can flood the network.

Note Currently, enabling this parameter breaks MTU path discovery and `traceroute` on an APX or a MAX TNT.

Preventing the redirection of UDP packets to the shelf controller

The `throttle-no-port-match-udp-traffic-on-slot` parameter in the `ip-global` profile prevents the TAOS unit from redirecting UDP packets (for UDP ports that are currently unknown to the TAOS unit) to the shelf controller. When two TAOS units are establishing the link for transmission of a VoIP call, both units do not always complete the call setup at the same time. However, a gateway starts sending UDP packets to the receiving gateway as soon its own call setup is complete. Enabling this parameter delays processing of these UDP packets until the link is fully established.

The `throttle-no-port-match-udp-traffic-on-slot` parameter accepts the following values:

Parameter	Setting
`throttle-no-port-match-udp-traffic-on-slot`	Set this parameter to either: • yes–Enables delaying processing of UDP packets for an unknown UDP port on a receiving gateway until call setup is completed on both APX units. **Note** This is the recommended setting for MultiVoice operations, to prevent overloading of the shelf controller when both gateways do not always complete the VoIP call setup at the same time. • no–Allows redirection of UDP packets for an unknown UDP port on a receiving gateway to its shelf controller. This is the default value. If the receiving gateway has not yet set up its port caches, the shelf controller receives the UDP packets until the call is fully set up.

Preventing premature fax/modem call time-outs

For H.323 VoIP, the `idle-timer` parameter in the `answer-defaults` profile prevents fax/modem calls from timing out prematurely. By default, once a fax/modem call is initiated at the near-end APX or MAX TNT, it only waits 120 seconds (2 minutes) for a response to the call request from the distant APX. When the near-end TAOS unit doesn't receive a response within that time, the call is dropped.

To ensure that these calls connect properly, the value of the `idle-timer` setting should be set to 0 (`idle-timer = 0`), causing the TAOS unit to wait without timing out for a response to the call request, as in the following example:

```
admin> read answer
ANSWER-DEFAULTS read
admin> list session
[in ANSWER-DEFAULTS:session-info]
call-filter = ""
data-filter = ""
filter-persistence = no
filter-required = no
fill-1 = 0
fill-2 = 0
idle-timer = 0
```

When Connection profiles are used, the value of the `idle-timer` setting should also be set to 0 (`idle-timer = 0`), as in the following example:

```
admin> read connection 1
CONNECTION/1 read
admin> list session
[in CONNECTION/1:session-options]
call-filter = ""
data-filter = ""
old-call-filter = 0
old-data-filter = 0
```

```
filter-persistence = no
filter-required = no
fill-1 = 0
idle-timer = 0
```

device-state profiles

When a DSP card enters the UP state for the first time, a `device-state` profile is created for each channel. When a DSP card is replaced by a different type of card or by the `slot -r` command, `device-state` profiles are deleted.

Implementing VoIP audio processing

An APX or MAX TNT performs centralized call processing of VoIP calls. The shelf controller handles the H.323 gateway administration, connection profile administration, and Dialed Number Identification Service (DNIS) call discrimination. The DSP card handles voice processing (such as codec processing, RTP/UDP processing, jitter buffer operation, echo cancellation, etc.).

VoIP audio routing is implemented as follows:

1 The caller dials a local access number for a MultiVoice Gateway.

2 The call is answered by a TAOS unit. This is the near-end TAOS unit.

   – If the TAOS unit is configured for two-stage dialing, the caller first hears a dial tone from the unit, then enters the destination telephone number. When callers are required to enter a Personal Identification Number (PIN), they enter their PIN at this first dial tone or prompt, then enter the destination telephone number when they hear a subsequent dial tone.

   – If the TAOS unit is configured for single-stage dialing, the caller dials the local access number and destination telephone number as a single string. When callers are required to enter a Personal Identification Number (PIN), they enter their PINs at a prompt from the TAOS unit.

3 The TAOS unit routes the call to a DSP on the MultiDSP card.

   – If the trunk connection from the PSTN/PBX supports DNIS signaling (T1/PRI or ISDN), the TAOS unit can use DNIS discrimination to identify VoIP calls.

   – If the trunk connection from the PSTN/PBX does not support DNIS signaling (T1 inband or R2), the TAOS unit uses the value assigned to the `default-call-type` parameter in the `call-route` profiles to identify VoIP calls.

4 When two-staged dialing is used, the DSP decodes DTMF tones and passes each number entered to the H.323 stack.

Note A MultiVoice Gateway automatically enables a-law companding upon detection of the DTMF by the decode DSP during call setup on a T1 card. As the call comes up on the T1, the DSP checks for a companding mode message that specifies a-law companding. When no message is sent, the DSP defaults to µ–law.

5 When the complete number is entered, the H.323 stack does a gateway lookup on the phone number to get the IP address of the far-end TAOS unit.

6 The H.323 stack negotiates with its peer on the far-end TAOS unit. Negotiations include exchanging UDP port numbers, frame mode, etc.

7 The far-end TAOS unit allocates a DSP on the MultiDSP card and places the outgoing call to the dialed telephone number.

8 The far-end TAOS unit informs the near-end TAOS unit when the call connects.

9 The callers proceed with the VoIP call.

Configuring VoIP call processing

Configuring VoIP call processing on the TAOS unit consists of two steps:

1 Creating the default `voip` profile (`voip { 0 0 }`)

2 Configuring the TAOS unit to process calls using IPDC or H.323 protocols

Creating the default voip profile

To configure or modify the VoIP call processing parameters, the default `voip` profile (`voip { 0 0 }`) must first be created on the TAOS unit. This is done using the `new` and `write` commands:

1 Use the `new` command to create the default `voip` profile, assigning `0 0` as the profile address:

```
admin> new voip { 0 0 }
VOIP/{ 0 0 } read
```

2 Use the `write` command to save the default profile:

```
admin> write
VOIP/{ 0 0 } written
```

Configuring routes for VoIP call processing

To configure call routes on an APX or a MAX TNT for VoIP call processing, you must perform the following tasks:

- Route VoIP calls from the PSTN to the MultiDSP cards for processing. This configuration includes the following:
  - Defining the `default-call-type` parameter for inbound call routing
  - Defining DNIS-specific trunk mappings
  - Modifying MultiDSP card `call-route` profiles to process voice or data calls
  - Defining preferred-source routing
- Route packetized voice across the IP network. This configuration includes the following:
  - Defining IP addressing schemes
  - Defining packet routing for H.323 VoIP calls
  - Configuring host routes or network routes for MultiVoice calls

Call routing parameters

TAOS supports simultaneous processing of VoIP and data calls. To simplify routing of VoIP and data traffic between the TAOS unit and PSTN for non-ISDN signaling trunks:

- Use the `default-call-type` parameter to identify inbound calls from a T1 or E1 trunk as VoIP or data for ingress call processing.
- Use DNIS-specific trunk mappings for routing of ingress VoIP calls.
- Process data and VoIP calls on different MultiDSP cards.
- Use preferred-source routing method (optional) for processing ingress data call types.
- Use trunk routing (optional) for processing egress VoIP calls.

For trunks using ISDN signalling, a TAOS unit ascertains the bearer capability (voice or data) of a call and uses that information to route the call to a modem (if a voice-service call) or HDLC channel (if a data call).

Using default-call-type for inbound call routing

When a T1 or E1 lines uses inband robbed-bit signaling, the `default-call-type` parameter in the `call-route` profile specifies the default call-type for incoming calls, for ingress call routing purposes:

Parameter value	Description
`digital`	Treat incoming calls as digital.
`voice`	Treat incoming calls as analog calls. Used for routing modem calls from the PSTN.
`dnis-or-voice`	Assign the call type based on the dialed number (must configure one or more DNIS profiles). Default to voice call type if there is no DNIS match.
`dnis-or-digital`	Assign the call type based on the dialed number (must configure one or more DNIS profiles). Default to digital call type if there is no DNIS match.
`voip`	Treat incoming calls as voice over IP.

When `default-call-type=voip`, all calls received on this trunk are processed as Voice over IP calls. When `default-call-type=digital`, all calls received on this trunk are processed as digital calls.

For example, if a TAOS unit is connected to a T1 line using inband signaling, the T1 profile contains the following:

```
[in T1/{ shelf-1 slot-1 2 }:line-interface]
signaling-mode=inband
robbed-bit-mode=inc-w-200
default-call-type=voip
```

As the call comes up from the T1 line, if the `default-call-type` for the associated T1 is set to `voip`, all calls received over that T1 are processed as VoIP calls. Calls associated with that trunk are routed by the shelf controller to the DSP card for processing.

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This same routing method is used when a TAOS unit is configured to process single-stage dialed numbers for H.323 VoIP calls. In this instance, multiple DNIS strings are collected by the TAOS unit. These multiple strings include:

- All or part of the TAOS unit gateway access number (this value is telephone company/provisioning dependent)
- The destination (called) telephone number

Note For trunks using ISDN signaling, use `default-call-type=voip` on a TAOS unit that processes only VoIP calls, and use `default-call-type=dnis-or-digital` on a TAOS unit that processes both VoIP and data calls.

To change the value of the `default-call-type` parameter, enter the commands as follows:

```
admin> read t1 {1 1 2}
T1/{ shelf-1 slot-1 2 } read
admin> set line-interface default-call-type = voip
admin> write
T1/{ shelf-1 slot-1 2 } written
```

In-bound calls received on all channels of this T1 line will be processed by the TAOS unit as VoIP calls.

Using DNIS-specific trunk mappings

The default `voip` profile, `voip { 0 0 }`, is a systemwide profile used for processing all VoIP calls. Additional `voip` profiles can be created to simplify processing and routing of VoIP calls. These user-defined `voip` profiles map incoming calls by identifying all calls associated with a specific DNIS string as VoIP calls.

Note The `default-call-type` parameter found in the T1 or E1 profiles should be set to `default-call-type=dnis-or-digital` when using DNIS-specific trunk mappings.

For example, if you created the following `voip` profiles, the TAOS unit processes all calls from the PSTN with these DNIS strings as VoIP calls.

```
admin> dir voip
   46  12/23/1998 09:48:55  { 0 0 }
   31  12/18/1998 09:50:06  { 8093190 0 }
   31  12/18/1998 10:07:16  { 8903190 0 }
```

The `voip-index` subprofile distinguishes between the default `voip` profile, `voip {0 0}` and any user-defined `voip` profiles:

```
admin> read voip { 8903190 0 }
VOIP/{ 8903190 0 } read
admin> list voip-index
[in VOIP/{ 8903190 0 }:voip-index]
gateway-access-number = 8903190
far-end-number = 0
```

The voip-index subprofile includes the following parameters:

Parameter	Setting
gateway-access-number	This is the telephone number dialed by a caller to access the TAOS unit. This telephone number is associated with a T1 trunk, which is used by the TAOS unit to receive in-bound calls from the PSTN. If the TAOS unit is configured to perform two-stage dialing of VoIP calls, this telephone number is dialed to access the TAOS unit.
far-end-number	This is the telephone number dialed by the TAOS unit to connect the call. This value should always be set to 0, since the caller normally enters the destination telephone number.

For DNIS-collecting trunks, a match is done on the DNIS to determine whether a call is a VoIP call. As the call comes from the T1 card, the supplied phone number is run through the current set of Voip profiles. When a match is found between this phone number and the gateway-access-number in the voip-index subprofile, the call is treated as a VoIP call.

Note Once the TAOS unit is initialized and these changes are committed, save the new configuration to flash memory or a TFTP server. The saved image can be retrieved to restore this configuration in the event that the TAOS unit must be re-initialized.

User-defined voip profiles are defined using the set, new, and write commands.

Using set and write commands to modify default voip profiles

To use the set and write commands, all the parameters in the default voip profile, voip { 0 0 }, must be set to their default values. The default voip profile is used to create user-defined voip profiles.

1 Use the read command to make the default voip profile the current working profile.

2 Use the list command to open the voip-index subprofile:

```
admin> list voip-index
[in VOIP/{ 0 0 }:voip-index]
gateway-access-number = 0
far-end-number = 0
```

3 Use the set command to assign the DNIS associated with a gateway access number on the TAOS unit to the gateway-access-number parameter:

```
admin> set gateway-access-number = 8903116
```

4 Use the write command to create the user-defined voip profile:

```
admin> write
VOIP/{ 8903116 0 } written
```

Note Using set and write does not work if you have already edited other parameters contained in the default voip profile.

Using new and write commands to create user-defined voip profiles

The new and write commands may be used, without restriction, to create user-defined voip profiles:

1 Use the read command to make the default voip profile the current working profile.

2 Use the new command to assign a profile for the gateway access number:

```
admin> new voip { 8903190 0 }
VOIP/{ 8903190 0 } read
```

3 Use the write command to save the new profile:

```
admin> write
VOIP/{ 8903190 0 } written
```

Processing VoIP and data calls on different MultiDSP slot cards

To enable the simultaneous processing of voice and data calls, you must create exclusive call routing types for each MultiDSP slot card. You create exclusive call-routing types by deleting call-route profiles for non-VoIP call types.

At startup, when a TAOS unit first detects the presence of a slot card, it automatically creates default call-route profiles to handle different call types. Software licenses on the shelf-controller that are enabled determine which type of call-route profiles are created. For each resource supported by the slot card (digital calls, PHS calls, VoIP calls), a separate call-route profile is created. The TAOS unit uses the presence or absence of a particular profile to filter calls that are accepted and processed by each MultiDSP card.

For each installed MultiDSP slot card, of call-route profiles can be created, these call-route profiles can be created as illustrated in the following callroute command output:

```
admin> callroute -d
device     #  source      type              tg  sa phone
1:03:01/0  1 0:00:00/0  digital-call-type   0   0
1:03:01/0  2 0:00:00/0  phs-call-type       0   0
1:03:01/0  3 0:00:00/0  voip-call-type      0   0
1:03:01/0  4 0:00:00/0  v110-call-type      0   0
1:03:01/0  4 0:00:00/0  voice-call-type     0   0
1:03:01/0  4 0:00:00/0  g729-call-type      0   0
1:03:01/0  4 0:00:00/0  g728-call-type      0   0
1:03:01/0  4 0:00:00/0  g723-call-type      0   0
1:03:01/0  4 0:00:00/0  rtfax-call-type     0   0
1:03:01/0  4 0:00:00/0  frgsm-call-type     0   0
```

The call-route-type parameter in the call-route profile identifies the type of resource (for example, digital-call-type, phs-call-type, voip-call-type). For each supported resource, a new call-route profile is created when the slot card is first installed.

The supported call-route types for the MultiDSP card include:

Table 2-2. Call-route types

Call-Route type	Description
any-call-type	Any type calls can be routed to a device with this call route type.
voice-call-type	Voice bearer calls, which do not include 3.1kHz audio call types or VoIP calls can be routed to a device with this call route type.
digital-call-type	General digital calls, including 3.1kHz audio bearer channel calls, routed to a host device can be routed to a device with this call route type.
trunk-call-type	Digital calls sent to a trunk device used for routing outbound calls to a particular trunk group can be routed to a device with this call route type.
phs-call-type	PHS calls can be routed to a device with this call route type.
v110-call-type	Digital calls recognized as containing V.110 rate-adapted bearer channels can be routed to a device with this call route type.
wormarq-call-type	Digital calls recognized as using WORM-ARQ technology for personal digital cellular phones can be routed to a device with this call route type. *WORM-ARQ is not currently supported on the APX 1000 platform.*
rtfax-call-type	When using the IPDC protocol, VoIP calls using T.38 fax can be routed to a device with this call route type.
g729-call-type	When using the IPDC protocol, VoIP calls using the G.729(A) codec can be routed to a device with this call route type.
g728-call-type	When using the IPDC protocol, VoIP calls using the G.728 codec can be routed to a device with this call route type.
g723-call-type	When using the IPDC protocol, VoIP calls using the G.723.1 codec can be routed to a device with this call route type.
frgsm-call-type	When using the IPDC protocol, VoIP calls using the Full-Rate GSM codec can be routed to a device with this call route type.
voip-call-type	VoIP calls using the G.711 codec and transparent fax and modem calls ("G.711 fallback") can be routed to a device with this call route type.

With codec-specific call-route types (for example, g729-call-type), more call-route profiles exist per slot card. For example, the 48-port model slot card has 11 call-route profiles because it supports 11 different types of resources (7 of the 11 call-route profiles are dedicated to audio codecs as listed in Table 2-4).

Use the callroute -ih command to view the call-route profiles created for each of the following MultiDSP slot cards:

- **48-port model**
  When a universal gateway has only the 48-port model slot card installed, the callroute-ih command displays these call-route profiles:

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```
admin> callroute -ih
   slot       #   cost  source        type            tg  sa  phone
 1:04:00/0    3    40   0:00:00/0  voice-call-type     0  0
 1:04:00/0    3    40   0:00:00/0  digital-call-type   0  0
 1:04:00/0    3    40   0:00:00/0  phs-call-type       0  0
 1:04:00/0    3    40   0:00:00/0  voip-call-type      0  0
 1:04:00/0    3    40   0:00:00/0  v110-call-type      0  0
 1:04:00/0    3    40   0:00:00/0  g729-call-type      0  0
 1:04:00/0    3    40   0:00:00/0  g728-call-type      0  0
 1:04:00/0    3    40   0:00:00/0  g723-call-type      0  0
 1:04:00/0    3    40   0:00:00/0  frgsm-call-type     0  0
```

- **96-port model.**
  When a universal gateway has only the 96-port model slot card installed, the `callroute-ih` command displays these `call-route` profiles:

```
admin> callr -ih
   slot       #   cost  source        type            tg  sa  phone
 1:05:00/0    3    30   0:00:00/0  voice-call-type     0  0
 1:05:00/0    3    30   0:00:00/0  digital-call-type   0  0
 1:05:00/0    3    30   0:00:00/0  phs-call-type       0  0
 1:05:00/0    3    30   0:00:00/0  voip-call-type      0  0
 1:05:00/0    3    30   0:00:00/0  v110-call-type      0  0
 1:05:00/0    3    30   0:00:00/0  g729-call-type      0  0
```

- **240-port model.**
  When a universal gateway has only the 240-port model slot card installed and has been hashed for voip only, the `callroute-ih` command displays these `call-route` profiles:

```
admin> callr -ih
   slot       #   cost  source        type            tg  sa  phone
 1:06:00/0    3    20   0:00:00/0  voice-call-type     0  0
 1:06:00/0    3    20   0:00:00/0  digital-call-type   0  0
 1:06:00/0    3    20   0:00:00/0  phs-call-type       0  0
 1:06:00/0    3    20   0:00:00/0  voip-call-type      0  0
 1:06:00/0    3    20   0:00:00/0  v110-call-type      0  0
 1:06:00/0    3    20   0:00:00/0  g729-call-type      0  0
```

- **288-port model.**
  When a universal gateway has only the 288-port model slot card installed and has been hashed for voip only, the `callroute-ih` command displays these `call-route` profiles:

```
admin> callr -ih
   slot       #   cost  source        type            tg  sa  phone
 1:07:00/0    3    20   0:00:00/0  voice-call-type     0  0
```

```
slot       #   cost  source      type            tg  sa phone

1:07:00/0  3   20   0:00:00/0   digital-call-type  0  0
1:07:00/0  3   20   0:00:00/0   phs-call-type      0  0
1:07:00/0  3   20   0:00:00/0   voip-call-type     0  0
1:07:00/0  3   20   0:00:00/0   v110-call-type     0  0
1:07:00/0  3   20   0:00:00/0   g729-call-type     0  0
```

- **288-port model configured for 480 ports.**
  When a universal gateway has only the 288-port model slot card configured to use 480 ports installed (madd3-voip-480), the callroute-ih command displays these call-route profiles:

  admin> **callr -ih**

```
   slot      #   cost  source      type           tg  sa phone
 1:05:00/0   3   10    0:00:00/0   voip-call-type  0   0
```

Note The call-route types for specific codecs (for example, g729-call-type) as defined in Table 2-2 are not supported when the universal gateways are under control of H.323 signaling. H.323 uses H.245 for codec negotiation between endpoints and therefore, call routes cannot specify codecs. For all codecs, only the voip-call-type call-route type can be used with H.323.

Depending upon whether you want the MultiDSP slot card to process VoIP or data calls, delete the call types as follows:

To process this call type	Delete call-route profiles
VoIP calls (voip-call-type)	any-call-type, digital-call-type, v110-call-type
Data calls (digital-call-type)	any-call-type, voip-call-type

For example, if you want a MultiDSP slot card to process only VoIP calls, delete the profiles when the call-route-type parameter is set to digital-call-type or v110-call-type, and any-call-type profiles for that MultiDSP slot card.

Note For all locations except Japan, the phs-call-type call-route profile need not be deleted for MultiDSP slot cards processing voice calls. PHS calls are only supported by PSTNs in Japan.

To remove call-route profiles, do the following:

1 Use the show command to identify all the MultiDSP (madd-card) slot cards installed in your TAOS unit:

```
admin>show
Shelf 1 ( standalone ):
{ shelf-1 slot-1 0 }        UP       8el-card
{ shelf-1 slot-2 0 }        UP       ether3-card
{ shelf-1 slot-3 0 }        UP       madd-card
{ shelf-1 slot-4 0 }        UP       madd-card
{ shelf-1 slot-5 0 }        UP       madd-card
```

```
{ shelf-1 slot-6 0 }       UP       madd-card
{ shelf-1 slot-7 0 }       UP       madd-card
{ shelf-1 slot-8 0 }       UP       madd-card
admin>
```

2 Delete the call-route profiles for each call type you do not want a MultiDSP slot card to accept. To delete the call-route profile for v110-call-type processing on the MultiDSP slot card in slot 3, enter the following command:

```
admin> delete call-route { { {1 3 0} 0} 4}
Delete profile CALL-ROUTE/{ { { shelf-1 slot-3 0 } 0 } 4 }? [y/n] y
CALL-ROUTE/{ { { shelf-1 slot-3 0 } 0 } 4 } deleted
admin>
```

Repeat this procedure for each call-route profile associated with an excluded call type.

Note Modification of the call-route profiles are made after the TAOS unit is initialized. Once the call-route profile changes are committed, save the new configuration to flash memory or a TFTP server. The saved image can be retrieved to restore this configuration in the event that a TAOS unit must be re-initialized.

Configuring preferred-source routing

Using preferred-source routing causes a TAOS unit to direct calls from the designated ingress device (such as, T1 or E1 slot cards) to a specific MultiDSP slot card. This can be used to limit the calls a MultiDSP slot card accepts for processing to a specific T1 or E1 channel and can also be used for routing data calls.

Preferred-source routing is implemented by assigning the address of a T1 or E1 channel to the preferred-source parameter in the call-route profiles for each data-call type configured for a MultiDSP card. The assigned address identifies the shelf, slot, and connection associated with a specific T1 or E1 trunk.

To configure preferred-source routing, enter the following:

1 Use the show command to identify all the T1 or E1 cards installed in your TAOS unit:

```
admin>show
Shelf 1 ( standalone ):
{ shelf-1 slot-1 0 }       UP       8e1-card
{ shelf-1 slot-2 0 }       UP       ether3-card
{ shelf-1 slot-3 0 }       UP       madd-card
{ shelf-1 slot-4 0 }       UP       madd-card
{ shelf-1 slot-5 0 }       UP       madd-card
{ shelf-1 slot-6 0 }       UP       madd-card
admin>
```

2 For each MultiDSP slot card, change the value assigned to the preferred-source parameter in its call-route profile for digital-call-type. To route calls received through any T1 or E1 line connected on slot 1 to the MultiDSP card in slot 4, enter the following command:

```
admin> read call-route { { {1 4 0} 0} 1}
CALL-ROUTE/{ { { shelf-1 slot-4 0 } 0 } 1 } read
admin>set preferred-source={{1 1 0} 0}
```

```
admin>write
CALL-ROUTE/{ { { shelf-1 slot-4 0 } 0 } 1 } written
admin>
```

You may configure a routing using all the T1 or E1 connections on the ingress card, as in the example, or specify an individual trunk by identifying a specific port on the ingress card, for example:

```
admin>set preferred-source={{1 1 4} 0}
```

Repeat this procedure until all T1 or E1 trunks are mapped to MultiDSP slot cards.

Note Make this modification after the TAOS unit is initialized. Once these changes are committed, save the new configuration to flash memory or a TFTP server. The saved image can be retrieved to restore this configuration in the event that an TAOS unit must be re-initialized.

Using automatic trunk routing (optional) for outbound voice calls

Trunk routing of outbound VoIP calls controls allocation of egress T1 or E1 trunks. A TAOS unit that connects a VoIP call to the destination telephone number can automatically route calls to the PSTN using a trunk group selected by the TAOS unit that initiated the call.

To utilize this trunk routing method:

- Trunk groups must be enabled on both TAOS units used to connect the call.
- Both TAOS units should have the same number of T1 or E1 trunks available for connecting VoIP calls.
- Both TAOS units must utilize the same trunk numbering scheme.

When trunk prefixing is enabled, the TAOS unit obtains the trunk group number of the ingress T1 trunk from the trunk-group setting in the T1 line profile, and prefixes it to the detected DNIS, the destination telephone number. The TAOS unit modifies the Q.931 Called Party Number information element (IE) in an H.225/Q.931 SETUP message, sending the DNIS number prefixed by the incoming trunk number to the TAOS unit which connects the voice call.

When the destination TAOS unit dials the call, it connects the call to the PSTN using a trunk assigned to the requested trunk group.

Enabling trunk groups

To enable automated trunk group processing of VoIP calls, you must configure the following:

Parameter	Setting
use-trunk-groups (system profile)	This parameter enables the use of trunk groups for all network lines. When this parameter is enabled (yes), all channels must be assigned a trunk group number for outgoing calls.

Parameter	Setting
num-digits-trunk-groups (system profile)	This parameter sets a limit of the number of digits (that is, 1-4) that may be used to designate trunk groups. The value assigned this parameter limits size of the values assigned to the trunk-group parameter to one through four place numbers.
trunk-group (t1 or e1 profile)	This parameter assigns a channel to a trunk group. In a t1 or e1 profile, the default is 9. Individual channels (1 - 9999) may be assigned to different trunk groups.
trunk-prefix-enable (voip profile)	This parameter enables outbound routing of VoIP calls over trunk groups designated by the ingress TAOS unit when set to yes. The ingress TAOS unit sends the trunk group address as part of the dial string for the destination telephone number.

Configuring egress call routing

A TAOS unit can be used as an egress MultiVoice Gateway, that is, used only for connecting calls to the switched telephone network, or as both an egress and an ingress gateway, both accepting calls from and connecting calls to the switched telephone network.

Configuring egress calls only

When egress calls use trunk group routing and a TAOS unit is used only as an egress MultiVoice Gateway, you control trunk group assignments by assigning a nonzero value to the trunk-group parameter in the call-route profile for a E1, T1, or T3 slot card. Using trunk-group=0 in the call-route profile configures the slot card for egress for any VoIP call routed for any trunk group. Since the MultiVoice Gateway is only processing outbound calls to the switched telephone network, assigning the trunk group at the slot level lets MultiVoice perform equal-cost routing of VoIP calls out to the switched telephone network.

For example, if a TAOS unit has three T1 slot cards installed in slot 11, slot 12 and slot 13, equal-cost routing of VoIP calls is achieved by assigning the T1 trunks on each slot to their own trunk group. For the T1 slot card in slot 11 of the TAOS unit, all eight T1 trunks are assigned to trunk group 11 using the trunk-group parameter in the call-route profile as follows:

```
tnt45> read call-route { { {1 11 0} 0 } 0 }
CALL-ROUTE/{ { { shelf-1 slot-11 0 } 0 } 0 } read

tnt45> list
[in CALL-ROUTE/{ { { shelf-1 slot-11 0 } 0 } 0 }]
index* = { { { shelf-1 slot-11 0 } 0 } 0 }
trunk-group = 11
phone-number = ""
preferred-source = { { any-shelf any-slot 0 } 0 }
call-route-type = trunk-call-type
```

Similarly, all T1 trunks on slot 12 would be assigned to trunk group 12, and all T1 trunks on slot 13 would be assigned to trunk group 13. When use-trunk-groups = yes in the system profile, the MultiVoice Access Manager can send instructions to the

TAOS unit to connect a call to the switched telephone network using trunk group 11, group 12, or group 13.

Configuring egress and ingress calls on T3 slot cards

When a TAOS unit is used as both an egress and ingress MultiVoice Gateway and T3 slot cards are used for the trunk connection to the switched telephone network, you assign trunk group numbers for the egress trunk group to the `trunk-group` parameter in the `t1:line-interface:channel-config:channel-config[n]` profile, and set `trunk-group=0` in the `call-route` profile of the T3 slot card.

When one T3 slot card provides both the ingress and egress connection to the switched telephone network, equal-cost routing of VoIP calls is achieved by assigning T1 trunks to different trunk groups. This provisions the TAOS unit to use different trunks on the T3 slot card for ingress and egress call processing.

For example, if a TAOS unit had a T3 slot card with 28 T1 trunks installed in slot 12, equal- cost routing of VoIP calls is achieved by assigning trunks 1 through 14 to trunk group 12 and trunks 15 through 28 to trunk group 13, at the DS0 level. For each DS0 in trunk group 12 set the `trunk-group` parameter in the `t1:line-interface:channel-config:channel-config[n]` profile as follows:

```
tnt45> list channel-config
[in T1/{ shelf-1 slot-12 1 }:line-interface:channel-config]

tnt45> list 1
[in T1/{ shelf-1 slot-12 1 }:line-interface:channel-config[1]]
channel-usage = switched-channel
trunk-group = 12
phone-number = ""
call-route-info = { any-shelf any-slot 0 }
```

When `use-trunk-groups` = yes in the `system` profile, the MultiVoice Access Manager can send instructions to the TAOS unit to connect a call to the switched telephone network using trunk group 12. All trunks in trunk group 13 would be available for processing ingress calls from the switched telephone network.

Configuring egress and ingress calls on several T1 or E1 slot cards

When a TAOS unit is used as both an egress and ingress MultiVoice Gateway and more than one T1 or E1 slot card is used for trunk connections to the switched telephone network, you assign trunk group numbers for the egress trunk group to the `trunk-group` parameters in both the `call-route` and the `t1:line-interface:channel-config:channel-config[n]` profiles to the egress trunk number. Equal-cost routing of VoIP calls is achieved by assigning the T1 trunks on each slot card to their own trunk groups.

For example, if a TAOS unit has two T1 slot cards installed in slot 12 and slot 13, all eight T1 trunks in slot 12 could be assigned to trunk group 12 as follows:

```
tnt45>list channel-config
[in T1/{ shelf-1 slot-12 1 }:line-interface:channel-config]

tnt45>list 1
[in T1/{ shelf-1 slot-12 1 }:line-interface:channel-config[1]]
channel-usage = switched-channel
trunk-group = 12
```

```
phone-number = ""
call-route-info = { any-shelf any-slot 0 }
```

Similarly, you assign the T1 slot card in slot 13 to trunk group 13 in its call-route profile. Each DS0 on the trunks on slot 13 would be assigned to trunk group 13. When use-trunk-groups = yes in the system profile, the MultiVoice Access Manager can send instructions to the TAOS unit to connect a call to the switched telephone network using either trunk group 12. The TAOS unit can accept ingress calls from the switched telephone network using trunk group 13.

Configuring egress and ingress calls on a single T1 or E1 slot card

When a TAOS unit is used as both an egress and ingress MultiVoice Gateway and only one T1 or E1 slot card is installed for trunk connections to the switched telephone network, you assign trunk group numbers for the egress trunk group to the Trunk-Group parameters in both the call-route and the t1:line-interface:channel-config:channel-config[n] profiles to the egress trunk number. Equal-cost routing of VoIP calls is achieved by assigning the T1 trunks on each slot card to their own trunk groups.

For example, T1 trunks 1 through 4 could be assigned to trunk group 90, and T1 trunks 5 through 8 to trunk group 91. In the call-route profile, use trunk-group = 90, as illustrated by the following example:

```
tnt45> list
[in CALL-ROUTE/{ { { shelf-1 slot-12 0 } 0 } 0 }]
index* = { { { shelf-1 slot-12 0 } 0 } 0 }
trunk-group = 90
phone-number = ""
preferred-source = { { any-shelf any-slot 0 } 0 }
call-route-type = trunk-call-type
```

When multiple trunk groups are used with trunks on a single E1 or T1 slot card, the value assigned to the trunk-group parameter in the call-route profile should match the value used for the first trunk group assigned to the DS0s connected to the slot card (such as, T1 trunks 1 through 4).

When use-trunk-groups = yes in the system profile, the MultiVoice Access Manager directs the TAOS unit to connect calls to the switched telephone network using the trunk group in the call-route profile, in this case, trunk group 90. The TAOS unit can accept ingress calls from the switched telephone network using trunk group 91.

Priority-based call routing

Priority-based call routing improves call-routing efficiency and flexibility when a single TAOS unit supports a mixture of different MultiDSP slot card models and each model supports different audio codecs.

To configure priority-based call routing, you set a parameter called cost in the appropriate call-route profile.

Priority based call routing

A single universal gateway can support a mixture of the following MultiDSP slot card models in a single chassis:

- 48-port model (TNTV-SL-ADI-C)

- 96-port model (APX8-SL-96DSP)
- 240-port model (APX-SL-DSP-3-L)
- 288-port model (APX-SL-DSP-3)

Note The 288-port model can be configured to use 480 ports for G.711, VoIP-only traffic (see"Configuring 480 ports for G.711-encoded VoIP-only calls" on page 2-65).

Using cost parameter settings in the call-route profile, calls can be routed to a slot card according to its cost value. The lower the cost value of a call route, the higher its priority for selection as the destination slot card for a call.

For example, each MultiDSP slot card model provides a different type of audio codec support. Suppose we have installed a 48-port slot card, a 96-port slot card, and a 288-port slot card in a universal gateway.

- The 48-port model supports these codecs: G.711, G.729(A), G.723.1, Full-rate GSM, Real-time Fax (T.38), and transparent fax/modem.
- The 96-port model and 288-port model support G.711, G729(A), Real-time Fax (T.38), and transparent fax/modem.

The 288-port model has more ports than the 96-port model, so if a G.711 call is placed, it should be routed to the 288-port model because it has the lowest cost value (see Table 2-3).

Default cost values for MultiDSP slot cards

The default cost value for each slot card can be changed by editing the cost parameter in the appropriate call-route profile for a particular resource (as defined by the call-route-type parameter).

The default cost parameters per MultiDSP slot card are as follows:

Table 2-3. Default cost values

MultiDSP slot card	Default cost of call route
48-port model	cost = 40
96-port model	cost = 30
288-port model	cost = 20
240-port model	cost = 20
480-port model (a 288-port slot card configured with 480 ports)	cost = 10

Note All VoIP call routes for a specific slot card initially activates with the above values.

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How calls are routed

For each audio codec, the following table illustrates how calls are routed when using the default cost value in the call-route profile:

Table 2-4. How Calls are Routed Using Cost Values

Audio codec	Slot card order (with lowest cost value first)
G.711	1 480-port model (a 288-port slot card configured with 480 ports).
	2 288-port model, 240-port model.
	3 96-port model.
	4 48-port model.
G.729(A)	1 288-port model, 240-port model.
	2 96-port model.
	3 48-port model.
Rt-Fax	1 288-port model, 240-port model.
	2 96-port model.
	3 48-port model.
G.728	1 48-port model.
G.723.1	1 48-port model.
Full-Rate GSM	1 48-port model.

Resetting the slot card

After making changes to any call-route profiles, verify that there are no active calls being processed by the slot card. Then reset the slot card or reset the entire universal gateway. To reset a slot card located in shelf 1, slot 8, proceed as follows:

1 **Check to see if the slot card is currently processing active calls:**

```
admin> modem -i | grep "1 8"
Modems allocated/in-use:
  Modem { 1 8 3 } ( Up Assign UP UP ENABLE )
```

2 **While waiting for active calls to be discontinued, prevent new calls from being routed into this slot card by disabling the modems:**

```
admin> mdmdisable 1 8
```

3 **Keep checking the status of the current calls until all calls are no longer being processed:**

```
admin> modem -i | grep "1 8"
```

You should see the following message before resetting the slot card:

```
Modems allocated/in-use:
  Modem { 1 8 3 } ( Up Assign UP UP ENABLE )
```

4 **Bring the slot down with the following command:**

```
admin> slot -d 1 8
```

5 **Remove the leftover profiles from the system with the following command:**

```
admin> slot -r 1 8
Slot 1/8, state change forced
```

6 **Activate the slot card with the following command:**

```
admin> slot -u 1 8
```

7 **To show status of the slot card, enter the following command:**

```
admin> sh
Controller { left-controller } (PRIMARY):
                         Reqd  Oper   Slot Type
{ shelf-1 slot-1 0}      UP    UP     8t1-card
{ shelf-1 slot-2 0 }     DOWN  RESET  ether3-card
{ shelf-1 slot-7 0 }     DOWN  RESET  t3-card
{ shelf-1 slot-8 0 }     UP    UP     madd3-voip-480
```

Configuring IP routing for H.323 call processing

There are two methods for implementing IP routing of MultiVoice RTP packets for H.323 VoIP call processing:

- Host routing method, which requires configuring static routes from each Ethernet interface on an TAOS unit to each Ethernet interface on all the other TAOS units in a network that can connect VoIP calls. This method is recommended for small networks, where the network does not provide core support for equal-cost multipath routing. This method requires:
  - Development of an IP addressing scheme
  - Creation of multiple static IP routes on each APX to provide equal-cost multipath routing to all the other APX systems that can connect VoIP calls
  - Defining trunk groups for egress call processing (optional)
- Network routing method, which requires configuring one default route from each Ethernet interface on a TAOS unit to a network switch or router that performs route management, regardless of how many other TAOS unit there are on the network. This method is recommended for networks, where the network provides core support for equal-cost multipath routing. This method requires:
  - Development of an IP addressing scheme
  - Creation of at least one IP route on each TAOS unit to the gateway router/ switch used by a TAOS unit to connect with the packet network.
  - Defining trunk groups for egress call processing (optional)

Depending upon the network design, using either the host routing or the network routing method enables equal-cost multipath routing of VoIP calls across the packet network. Equal-cost multipath routing maximizes use of available Ethernet interfaces, packet network channels, etc., to route VoIP calls across the aggregated bandwidth of the packet network between MultiVoice Gateways.

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IP addressing schemes

An appropriate IP addressing scheme is required on each TAOS unit to ensure that VoIP packets are properly routed across the packet network. The MultiVoice network administrators should adhere to the following rules when assigning IP addresses to TAOS network interfaces:

- The soft IP address should be unique.
- The IP addresses assigned to individual Ethernet interfaces on shelf controllers and slot cards should be unique.
- The `system-ip-addr` parameter address and Ethernet 3 card IP address should be on different logical subnets.

Following those configuration rules aids in routing VoIP call data through the Ethernet card interfaces of a TAOS unit, rather then the shelf controllers. Sending call data directly to the shelf controllers causes calls to be rejected, generating a log message similar to the following:

```
LOG warning, Shelf 1, Slot 13, Time: 10:35:25--
[1/13/93/0] Invalid RTP path (madd->shelf) [MBID 470]
[V:aa239039-f22c-f002-1d31-0d9dc]
```

Packet routing for H.323 VoIP calls

When two TAOS units involved in an H.323 VoIP call are setting up the VoIP packet connection via H.323 signaling, they use a destination address value equal to the destination's `system-ip-addr` setting to send voice packets to each other. Proper IP address assignment and route setup are necessary to force RTP packets generated on the DSP card to go directly to an Ethernet slot card rather than the shelf controller. When assigning IP addresses, the following rules apply for all configurations to ensure that VoIP packet routing bypasses the shelf controller:

- Assign the IP addresses on the Ethernet cards in a TAOS unit to logically different subnets.
- Assign the IP address in the `system-ip-addr` parameter on a TAOS unit to a logically different subnet from any of the Ethernet slot card IP addresses.
- Enable IP Route and IP Port caching in the `ip-global` profile. For example:

```
admin> read ip-global
IP-GLOBAL read

admin> set iproute-cache-enable = yes
admin> set iproute-cache-size = 0 (0 = unlimited size)
admin> set ipport-cache-enable = yes
admin> write
IP-GLOBAL written
```

Configuring host routes

Host route configuration is simple because it doesn't require a network core that supports equal-cost multipath routing. However, host routing requires a more complex configuration on the TAOS unit. Host routing is typically used for environments with a small number of gateways (for example, demo and test environments) since this configuration doesn't use intermediate routers to provide equal-cost multipath routing across the IP network.

The host route method of configuration establishes a host route for every possible destination MultiVoice Gateway for each Ethernet slot card in that TAOS unit. While this can become unwieldy for a network with many MultiVoice Gateways, it simplifies the IP network configurations for networks using few MultiVoice Gateways.

For example, the following tables illustrate the IP addressing scheme for a TAOS unit-173 MultiVoice Gateway, which allows it to route VoIP calls to the TAOS unit-186 MultiVoice Gateway. This is a "back-to-back" configuration.

The IP addresses in Table 2-5 are used as logical network gateways by TAOS unit-173 for sending and receiving VoIP call data across the IP network. The logical network gateways are associated with network ports on the Ethernet-3 slot cards that provide the physical network connection from TAOS unit-173 to the packet network.

Table 2-5. IP address table for TAOS unit-*173*

System IP address	Ethernet interfaces	IP address
192.168.35.173	{{0 0 0}0}*	192.168.35.173/24
	{{1 3 1}0}	208.168.25.173/24
	{{1 4 1}0}	208.168.15.173/24

* This is the soft IP address by which the TAOS unit is known to the network.

TAOS unit-173 uses the IP addresses in Table 2-6 to define the destination IP address for sending VoIP call data across the network to TAOS unit-196. The host routes are configured by creating two IP Route profiles to TAOS unit-196 on TAOS unit-173. In each IP Route profile, the system IP address for TAOS unit-196 is assigned to the Dest-Address parameter and one of the IP addresses associated with ports on the Ethernet-3 cards in TAOS unit-196, that provide the physical network connection to the packet network, is assigned to the `gateway-address` parameter.

Table 2-6. IP Routing table for host routes from TAOS unit-*173 to* TAOS unit-*196*

dest-address parameter*	gateway-address parameter†
192.168.35.196/32	208.168.25.196
192.168.35.196/32	208.168.15.196

* Using the 32-bit subnet address fully qualifies the IP address, making this a host route.
† This is the Ethernet port IP address of the destination TAOS unit.

The first `ip-route` profile would be similar to the following:

```
admin> list
[in IP-ROUTE/tnt_196_1]
name* = tnt_196_1
dest-address = 192.168.35.196/32
gateway-address = 208.168.25.196
```